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SIEDLER, DOROTHY S	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/685,586	LIU ET AL.	
	Examiner	**Art Unit**	
	Dorothy Sarah Siedler	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 12-28-07.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-31 is/are pending in the application.

4a) Of the above claim(s) 3,13,20,25 and 31 is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1,2,4-12,14-19,21-24 and 26-30 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date 2-1-08,2-13-08,4-9-07.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

The examiner notes applicant's request to consider EP-1 422 692 A2, JP-361285570A and U.S. Patent 6,748,350. EP-1 422 692 A2 and JP-361285570A, which are on the 1449 dated April 9, 2007, were overlooked, but have now been considered. The applicant will receive an updated 1449 with this communication. The examiner also notes that U.S. patent 6,748,350, filled on the July 2, 2007 IDS, is titled as, "Method to compensate for stress between heat spreader and thermal interface material", and has no bearing on the subject matter of the instant application; the U.S. patent was crossed out on the 1449, since it cannot be considered prior art.

Applicant's arguments with respect to the objection to the specification and the 1122nd paragraph rejection of claim 6 are persuasive, therefore the objection and rejection is withdrawn.

Applicant also argues that, "KUBALA and SIEGLER, do not disclose or suggest segmenting an input audio stream into predetermined length intervals such that portions of the intervals overlap one another, as recited in amended claim 1", concluding that, "KUBALA does not disclose that the input frames are predetermined length intervals. KUBALA does not disclose or suggest anything about the length of the input frames. Therefore, KUBALA cannot disclose or suggest segmenting the input audio stream into predetermined intervals such that portions of the intervals overlap one another, as recited in amended claim 1" (Remarks page 12 and 13); however the examiner

respectfully disagrees. ***Kubala*** performs speaker segmentation of input speech, where frames (intervals) are initially labeled as speech or non-speech. In any speech processing system, input speech cannot be segmented into frames without first defining the length of that frame (interval). Since the system of ***Kubala*** processes speech frames (intervals), it is therefore inherent that the frames were first created as predetermined intervals of the incoming speech.

Applicant's arguments with respect to the rejection of claim 3 using ***Siegler***, now incorporated into claim 1, have been considered but are moot in view of the new ground(s) of rejection.

Claim Objections

Claim 26 objected to because of the following informalities: Claim 26 is dependent from cancelled claim 25. The examiner considers this a typographical error, and therefore interprets claim 26 as dependent from independent claim 23. Appropriate correction is required.

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 1, 7, 11, 14, 23, 24 and 30 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Kubala*** ("Integrated Technologies for Indexing Spoken Language" ACM 2000) in view of ***Aversano*** ("A New Text-Independent Method for Phoneme Segmentation" IEEE 2001).

As per amended claim 1, ***Kubala*** discloses a method for detecting speaker changes in an input audio stream comprising:

Segmenting the input audio stream into predetermined length intervals (page 53, first paragraph, *the speech is input as frames*);

Decoding the intervals to produce a set of phones corresponding to each of the intervals (page 53, *phone class recognition is performed on each frame. Therefore it is inherent that a set of phones was decoded for each frame*);

Generating a similarity measurement based on a first portion of the audio stream that is within one of the intervals and that occurs prior to a boundary between adjacent phones in one of the intervals and a second portion of the audio stream that is within the one of the intervals and that occurs after the boundary (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*; and

Detecting speaker changes based on the similarity measurement (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*; and

Outputting an indication of the detected speaker changes (page 49, *the system outputs a summary which indicates the location of speaker changes within the recording*).

Kubala does not disclose segmenting the input stream into predetermined intervals such that the portions of the intervals overlap one another. However, segmenting speech into overlapping intervals is known, as indicated in ***Aversano***. ***Aversano*** discloses a system that segments an input speech signal into 20 ms frames (intervals), with 10ms of frame-overlap (page 516, section 2, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of segmenting an input stream into overlapping predetermined intervals in ***Kubala***, since it would produce the predictable result of improving the detection of speech, including speaker, changes at the boundaries of the frames (intervals).

As per original claim 7, ***Kubala*** in view of ***Aversano*** disclose the method of claim 1, and ***Kubala*** further discloses wherein the decoded set of phones is selected from a simplified corpus of phone classes (page 53, first paragraph, *speech is labeled using speech and non-speech models*).

As per amended claim 11, ***Kubala*** discloses a device for detecting speaker changes in an audio signal, the device comprising:

A processor (page 52, second column, first paragraph, *the Pentium II processor*);

A memory (page 52, *the system is run on a computer using a Pentium II processor, therefore it is inherent that there are instruction stored in memory*) containing instructions that when executed by the processor cause the processor to:

Segment the audio signal into predetermined length intervals (page 53, first paragraph, *the speech is input as frames*),

Decode the intervals to produce a set of phones corresponding to each of the intervals (page 53, *phone class recognition is performed on each frame. Therefore it is inherent that a set of phones was decoded for each frame*),

Generate a similarity measurement based on a first portion of the audio signal that occurs prior to a boundary between phones in one of the sets of phones of an interval and a second portion of the audio signal that occurs after the boundary (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*, and

Detect speaker changes based on the similarity measurement boundary (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*, and

Store an indication of the detected speaker changes (page 49 and 52, and Figure 5, *the structural summarization, including the locations of speakers in a recording, are stored in an XML file in the Indexer subsystem*).

Kubala does not disclose segmenting the input stream into predetermined intervals such that the portions of the intervals overlap one another. However, segmenting speech into overlapping intervals is known, as indicated in ***Aversano***. ***Aversano*** discloses a system that segments an input speech signal into 20 ms frames (intervals), with 10ms of frame-overlap (page 516, section 2, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of segmenting an input stream into overlapping predetermined intervals in ***Kubala***, since it would produce the predictable result of improving the detection of speech, including speaker, changes at the boundaries of the frames (intervals).

As per original claim 14, this claim contains limitations that are similar to the limitations cited in claim 7, and is rejected for similar reasons.

As per previously presented claim 23, ***Kubala*** discloses a system comprising:

An indexer configured to receive input audio data and generate a rich transcription from the audio data, the rich transcription including metadata that defines

speaker changes in the audio data (page 49 and 52, and Figure 5, *the Indexer subsystem creates a structural summarization, including a transcript indicating locations of speakers in a recording*), the indexer including:

A segmentation component configured to divide the audio into segments of a predetermined length (page 53, first paragraph, *the speech is input as frames*),

A speaker change detection component configured to detect locations of speaker changes in the audio data based on a similarity value calculated at locations in the segments that correspond to phone class boundaries (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score))*;

A memory system for storing the rich transcription (page 49 and 52, and Figure 5, *the structural summarization, including the locations of speakers in a recording, is stored in an XML file in the Indexer subsystem*); and

A server configured to receive requests for documents and to respond to the requests by transmitting ones of the rich transcriptions that match the requests (page 55, Information Retrieval, *information indexing and retrieval take place on the Rougn'n'Ready server*).

Kubala does not disclose a segmentation component configured to divide the audio data into overlapping segments of a predetermined length. However, segmenting speech into overlapping intervals is known, as indicated in ***Aversano***. ***Aversano***

discloses a system that segments an input speech signal into 20 ms frames (intervals), with 10ms of frame-overlap (page 516, section 2, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of segmenting an input stream into overlapping predetermined intervals in ***Kubala***, since it would produce the predictable result of improving the detection of speech, including speaker, changes at the boundaries of the frames (intervals).

As per original claim 24, ***Kubala*** in view of ***Aversano*** disclose system of claim 23, and ***Kubala*** further discloses wherein the indexer further includes at least one of: a speaker clustering component, a speaker identification component, a name spotting component, and a topic classification component (pages 52-55 and Figure 5).

As per amended claim 30, ***Kubala*** discloses a device comprising:

Means for segmenting the input audio stream into predetermined length intervals (page 53, first paragraph, *the speech is input as frames*);

Means for decoding the intervals to produce a set of phones corresponding to each of the intervals (page 53, *phone class recognition is performed on each frame. Therefore it is inherent that a set of phones was decoded for each frame*);

Means for generating a similarity measurement based on audio within one of the intervals that is prior to a boundary between adjacent phones and based on audio within the one of the intervals that is after the boundary (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*;

Means for detecting speaker changes based on the similarity measurement (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*; and

Means for outputting the detected speaker changes (page 49 and 52, and Figure 5, *the structural summarization, including the locations of speakers in a recording, are stored in an XML file in the Indexer subsystem*).

Kubala does not disclose segmenting the input stream into predetermined intervals such that the portions of the intervals overlap one another. However, segmenting speech into overlapping intervals is known, as indicated in ***Aversano***. ***Aversano*** discloses a system that segments an input speech signal into 20 ms frames (intervals), with 10ms of frame-overlap (page 516, section 2, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of segmenting an input stream into overlapping predetermined intervals in ***Kubala***, since it would produce the predictable result of improving the detection of speech, or speaker, changes at the boundaries of the frames (intervals).

Claims 2,12 and 26 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Kubala*** in view of ***Aversano,*** and further in view of ***Beigi*** ("A Distance Measure Between Collection of Distributions and it's Application to Speaker Recognition" IEEE 1998).

As per original claim 2, ***Kubala*** in view of ***Aversano*** disclose the method of claim 1, however neither disclose wherein the predetermined length intervals are approximately thirty seconds in length. ***Beigi*** discloses the use of intervals that are approximately thirty seconds long in a speech recognition system (page 756, Section 4 Results, *30 seconds of speech is used for training*). ***Beigi*** discloses a system that models speech for different speakers as two sets of statistical distributions. A meaningful distance measure between the two distributions is calculated, which can then be used for speaker classification, speech segmentation and speaker verification. ***Beigi*** also uses thirty seconds of speech data as enrollment data. Therefore, the examiner argues that it is old and well known to segment audio into predetermined intervals approximately thirty seconds long.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to use predetermined length intervals approximately thirty seconds in length in ***Kubala***, since an interval of that length would provide robust data for training a speaker segmentation model.

As per original claim 12, this claim contains limitations that are similar to the limitations cited in claim 2, and is rejected for similar reasons.

As per original claim 26, this claim contains limitations that are similar to the limitations cited in claim 2, and is rejected for similar reasons.

Claims 4-6, 8-10, 15-18, 21,22, 27-29 are rejected under 35 U.S.C. 103(a) as being unpatentable over ***Kubala*** in view of ***Aversano,*** and further in view of ***Liu*** ("Fast Speaker Change Detection for Broadcast News Transcription and Indexing" 1999).

As per previously presented claim 4, ***Kubala*** in view of ***Aversano*** disclose the method of claim 1, however neither disclose wherein generating a similarity measurement includes: calculating cepstral vectors for the audio stream prior to the boundary and after the boundary, and comparing the cepstral vectors. ***Liu*** discloses a system for speaker change detection that calculates cepstral vectors for the audio stream prior to the boundary and after the boundary, and compares the cepstral vectors (Section 4 Speaker Change Detection, subsection Distance Measure Criterion, *cepstral vectors are used in the distance measure (similarity measure)*). In addition, cepstral vectors are

one of many vector types used to represent speech features in speech processing tasks.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to calculate cepstral feature vectors for the audio stream prior to and after the boundary in ***Kubala*** and ***Aversano***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp to achieve the predictable result of calculating robust and reliable feature vectors.

As per original claim 5, ***Kubala*** in view of ***Aversano,*** further in view of ***Liu*** disclose the method of claim 4, however neither ***Kubala*** nor ***Aversano*** disclose wherein the cepstral vectors are compared using a generalized likelihood ratio test. However, ***Kubala*** does disclose calculating a similarity measure using the Likelihood ratio test. In addition, ***Liu*** discloses wherein the cepstral vectors are compared using a generalized likelihood ratio test (Section 4 Speaker Change Detection, subsection Distance Measure Criterion). Cepstral vectors are one of many vector types used to represent speech features in speech processing tasks.

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to compare the cepstral vectors using a generalized likelihood ratio test in ***Kubala*** and ***Aversano***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp to achieve the result of calculating robust and reliable feature vectors.

As per original claim 6, ***Kubala*** in view of ***Aversano,*** further in view of ***Liu*** disclose the method of claim 5, but neither ***Kubala*** nor ***Aversano*** disclose wherein a speaker change is detected when the generalized likelihood ratio test produces a value less than a preset threshold. However, the generalized likelihood ratio test is a standard method used to detect abrupt changes in a non-stationary signal, where an optimized likelihood forms a decision function that is compared to a threshold; a change point is indicated when the value exceeds a threshold. ***Liu*** discloses a system that uses the GLR (Section 4 Speaker Change Detection, subsection the critical region). The generalized likelihood function used in the instant application is simply the reciprocal of the GLR commonly used, however its function is the same, i.e. indicating a change point in a non-stationary signal.

Therefore it would have been obvious to one of ordinary in the art at the time of the invention to use the generalized likelihood ratio test to indicate a change when the value is less than a threshold in ***Kubala*** and ***Aversano***, since one of ordinary skill in the art has good reason to pursue the options within his or her technical grasp in order to make a reliable decision as to when a speaker change has occurred in input audio data.

As per original claim 8, ***Kubala*** in view of ***Aversano,*** disclose the method of claim 7, however neither disclose wherein the simplified corpus of phone classes includes a phone class for vowels and nasals, a phone class for fricatives, and a phone class for

obstruents. ***Liu*** discloses wherein the simplified corpus of phone classes includes a phone class for vowels, nasals, fricatives and obstruents (Section 3 Phone-Class Decode, Figure 1).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to have corpus of phone classes that include a phone class for vowels and nasals, a phone class for fricatives, and a phone class for obstruents in ***Kubala*** and ***Aversano***, since vowels and nasals are similar in that they both have pitch and high energy, and can therefore be combined to significantly speed up processing, as indicated in ***Liu*** (section 3).

As per original claim 9, ***Kubala*** in view of ***Aversano,*** further in view of ***Liu*** disclose the method of claim 8, and ***Kubala*** further discloses wherein the simplified corpus of phone classes further includes a phone class for music, laughter, breath and lip-smack (page 53, first paragraph). However, neither ***Kubala*** nor ***Aversano*** explicitly disclose wherein the simplified corpus of phone classes further includes a phone class for silence. ***Liu*** discloses wherein the simplified corpus of phone classes further includes a phone class for silence (section 3).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to have a phone class for silence in ***Kubala*** and ***Aversano***, since non-speech events poses valuable information about speaker changes, and can be used to more accurately determine speaker changes, as indicated in ***Liu*** (section 3, first paragraph).

As per original claim 10, ***Kubala*** in view of ***Aversano*** disclose the method of claim 7, however neither disclose wherein the simplified corpus of phone classes includes approximately seven phone classes. ***Liu*** discloses wherein the simplified corpus of phone classes includes approximately seven phone classes (section 3).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to have a corpus of phone classes including approximately seven phone classes in ***Kubala*** and ***Aversano***, since it reduces the number of active nodes during decoding, and thus speeds up processing, as indicated in ***Liu*** (section 3, last paragraph).

As per original claim 15, this claim has limitations similar to claim 8, and is therefore rejected for similar reasons.

As per original claim 16, this claim has limitations similar to claim 9, and is therefore rejected for similar reasons.

As per original claim 17, this claim has limitations similar to claim 10, and is therefore rejected for similar reasons.

As per amended claim 18, ***Kubala*** discloses a device for detecting speaker changes in an audio signal, the device comprising:

A segmentation component configured to segment the audio signal into predetermined length intervals (page 53, first paragraph, *the speech is input as frames*);

A phone classification decode component configured to decode the intervals to produce a set of phone classes corresponding to each of the intervals (page 53, *phone class recognition is performed on each frame. Therefore it is inherent that a set of phones was decoded for each frame*); and

A speaker change detection component configured to detect locations of speaker changes in the audio signal based on a similarity value calculated over a first portion of the audio signal that occurs prior to a boundary between phone classes in one of the intervals and a second portion of the audio signal that occurs after the boundary in the one of the intervals boundary (page 53, second paragraph, *speaker change is hypothesized at every phone boundary using a form of a likelihood ratio test (similarity score)*;

Wherein an indication of the detected locations of speaker changes are output from the device (page 49, *the system outputs a summary which indicates the location of speaker changes within the recording*).

Kubala does not disclose segmenting the input stream into predetermined intervals such that the portions of the intervals overlap one another. However, segmenting speech into overlapping intervals is known, as indicated in ***Aversano***.

Aversano discloses a system that segments an input speech signal into 20 ms frames (intervals), with 10ms of frame-overlap (page 516, section 2, second paragraph).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to apply the known technique of segmenting an input stream into overlapping predetermined intervals in ***Kubala***, since it would produce the predictable result of improving the detection of speech, or speaker, changes at the boundaries of the frames (intervals).

Kubala also does not disclose wherein a number of possible phone classes being approximately seven. ***Liu*** discloses wherein the simplified corpus of phone classes includes approximately seven phone classes (section 3).

Therefore it would have been obvious to one of ordinary skill in the art at the time of the invention to have a corpus of phone classes including approximately seven phone classes in ***Kubala*** and ***Aversano***, since it reduces the number of active nodes during decoding, and thus speeds up processing, as indicated in ***Liu*** (section 3, last paragraph).

As per original claim 21, this claim contains limitations similar to those on claim 8, and is therefore rejected for similar reasons.

As per original claim 22, this claim contains limitations similar to those on claim 9, and is therefore rejected for similar reasons.

As per original claim 27, this claim contains limitations similar to those on claim 8, and is therefore rejected for similar reasons.

As per original claim 28, this claim contains limitations similar to those on claim 9, and is therefore rejected for similar reasons.

As per original claim 29, this claim contains limitations similar to those on claim 10, and is therefore rejected for similar reasons.

Claims 19 is rejected under 35 U.S.C. 103(a) as being unpatentable over ***Kubala*** in view of ***Aversano***, in view of ***Liu*** as applied to claim 18 above, and further in view of ***Beigi***.

As per claim 19, this claim contains limitations similar to those on claims 2 and 12, and is therefore rejected for similar reasons.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Dorothy Sarah Siedler whose telephone number is 571-270-1067. The examiner can normally be reached on Mon-Thur 9:30am-5:30pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on 571-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

DSS

3/26/2008

/Talivaldis Ivars Smits/
Primary Examiner, Art Unit 2626